CLAIMS

1. An audio signal processing method comprising the steps of:

supplying an audio signal to each of a plurality of digital filters;

supplying outputs from the plurality of digital filters to each of a plurality of speakers forming a speaker array to form a sound field;

setting a predetermined delay time to be given in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the digital filters and each of the speakers will coincide with each other; and

adjusting the amplitude characteristic of the plurality of digital filters so that a low-pass filter characteristic will be given to the synthesis response of the audio signal at a second point in the sound field.

2. The audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach at least one of the first and second point after it is reflected by a wall surface.

3. The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the latter.

4. The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of

digital filters is read from a data base and set for each of the plurality of digital filters.

5. The audio signal processing method according to claim 1, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time is over-sampled for a shorter period than the sampling period to provide a sample train and the sample train is down-sampled to provide pulse-waveform data of the sampling period; and

the factor data is set for a part to be delayed by the digital filters on the basis of the pulse-waveform data.

6. The audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time, which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of the predetermined delay time, which includes the decimal part by the digital filters.

7. The audio signal processing method according to claim 5, wherein:

the over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

8. The audio signal processing method according to claim 7, wherein:

the pulse-waveform data to be delayed by a delay time which is m/N (m = 1 to N − 1) of the sampling period is pre-stored in a data base; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the digital filters.

9. The audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the digital filters.

10. An audio signal processor comprising a plurality of digital filters each supplied with an audio signal, wherein

each of the plurality of digital filters supplies to each of a plurality of speakers forming a speaker array to form a sound field;

each of the plurality of digital filters has a predetermined delay time so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the digital filters and each of the speakers will coincide with each other; and

each of the plurality of digital filters has an amplitude characteristic so that a low-pass filter characteristic will be given to the synthesis response of the audio signal at a second point in the sound field.

11. The audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second point after it is

reflected by a wall surface.

12. The audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the latter.

13. The audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters.

14. The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal,

there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the digital filters.

15. The audio signal processor according to claim 14, wherein:

the over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

16. The audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the digital filters.

17. The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the digital filters.

18. The audio signal processor according to claim 17, wherein:

the over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral

multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

19. The audio signal processor according to claim 17, wherein:

a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the digital filters.

20. The audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the digital filters.